IN THE CLAIMS

Please amend the claims to read as follows:

1. (Currently Amended) A system for throttling network packets generated from an audio signal, comprising:

an encoder that encodes the audio signal into audio packets;

an interface buffer for storing the audio packets; ~~and~~

~~a processor that converts~~ multiple processors that convert the audio packets into the network packets;

~~, and~~ a monitor that monitors available utilization capacity for at least one of the interface buffer and the ~~processor~~ multiple processors ;

the ~~processor~~ monitor ~~increasing~~ varying a percentage of the multiple processors that increase an amount the audio signal that the encoder encodes into the audio packets when the available utilization capacity indicates high utilization for at least one of the interface buffer or the ~~processor~~ multiple processors.

2. (Currently Amended) A gateway for throttling network packets generated from an audio signal, comprising:

an encoder that encodes the audio signal into audio packets;

~~a processor that converts~~ multiple processors that convert the audio packets into the network packets, ~~the processor~~ a percentage of the multiple processors increasing an amount of the audio signal that the encoder encodes into the audio packets varied according to an ~~when~~ available utilization capacity of the gateway ~~drops~~ dropping below a low availability threshold; and

a buffer having a free queue that receives the audio packets from the encoder, the available utilization capacity of the gateway varying according to space available in the free queue for receiving the audio packets.

3. (Canceled)

4. (Canceled)

5. (Previously Presented) A system according to claim 1 wherein the utilization capacity varies according to a number of audio signals from incoming calls the system is currently converting into network packets.

6. (Currently Amended) ~~A system according to claim 1 including~~ A system for throttling network packets generated from an audio signal, comprising:

multiple encoders each encoding audio signals into audio packets ~~for a different incoming call;~~

an interface buffer for storing the audio packets; and

a processor monitoring available utilization capacity for the interface buffer and the multiple encoders and ~~, the processor~~ varying a percentage of the encoders that increase ~~the~~ an audio packet size according to the utilization capacity.

7. (Previously Presented) A system according to claim 1 wherein the encoder encodes about 20 milliseconds of the audio signal into the audio packets when the available utilization capacity is greater than a first threshold, encodes about 40 milliseconds of the audio signal into the audio packets when the available utilization capacity falls below the first threshold, and encodes "more than >60 60 milliseconds" of the audio signal into the audio packets when the available utilization capacity falls below a second threshold less than the first threshold.

8. (Previously Presented) A system according to claim 1 wherein the audio signal is received over an incoming Public Services Telephone Network (PSTN) call and the network packets are IP packets transferred out over an IP network.

9. (Currently Amended) A method for throttling network packets in a voice gateway, comprising:

encoding ~~an~~ audio ~~signal~~ signals;

formatting the encoded audio ~~signal~~ signals into Voice Over Internet Protocol (VoIP) packets using ~~a~~ multiple central processing ~~unit~~ units;

storing the VoIP packets in an interface buffer;

monitoring utilization of at least one of the interface buffer and the multiple central processing ~~unit~~ units; ~~and~~

controlling size of the VoIP packets by increasing a number of samples of the encoded audio ~~signal~~ signals in the VoIP packets when the monitored utilization indicates

high interface buffer utilization or high utilization of the central processing ~~utilization~~ units; and

varying a percentage of the multiple central processing units used for increasing the number of samples in the VoIP packets according to the monitored utilization.

10. (Currently Amended) A method according to claim 9 including formatting the encoded audio signals using the multiple central processing ~~unit~~ units and varying the VoIP packet size according an amount of processing capacity of the multiple central processing ~~unit~~ units used for formatting the encoded audio ~~signal~~ signals into VoIP packets.

11. (Original) A method according to claim 10 including:

storing the VoIP packets in the interface buffer before transmitting the VoIP packets over a VoIP network;

monitoring the interface buffer by determining an amount of free space in the interface buffer currently not storing VoIP packets; and

controlling the VoIP packet size according to the amount of free space currently in the interface buffer.

12. (Currently Amended) A method according to claim 11 including periodically monitoring the amount of free space in the interface buffer and the available processing capacity of the central processing ~~unit~~ units and controlling the VoIP packet size according to that periodic monitoring.

13. (Previously Presented) A method for throttling network packets in a voice gateway, comprising:

using multiple digital signal processors to encode multiple audio signals at the same time;

formatting the encoded audio signal into Voice Over Internet Protocol (VoIP) packets using a central processing unit;

storing the VoIP packets in an interface buffer;

monitoring utilization of at least one of the interface buffer and the central processing

unit; and

varying a percentage of the digital signal processors that increase the VoIP packet size according to the monitored utilization.

14. (Previously Presented) A method according to claim 13 including:

varying a percentage of the digital signal processors that increase the VoIP packet size according to the amount of free space in the interface buffer and an amount of processing capacity of the central processing unit used for switching the encoded audio signal to the IP network.

15. (Canceled)

16. (Canceled)

17. (Previously Presented) A computer program for use with a network processing device, said computer program, comprising:

a processor load monitor that monitors utilization of a processor in the network processing device;

a throttle indicator that generates a throttle value according to the monitored processor utilization, the throttle value used by the network processing device to vary an amount of an audio signal that is encoded into the audio packets; and

wherein size of the audio packets are throttled in a percentage of multiple digital signal processors wherein the percentage is proportional to the throttle value.

18. (Canceled)

19. (Currently Amended) A computer program according to claim ~~18~~ 17 wherein the ~~number~~ amount of ~~samples of~~ the audio signal encoded in the audio packets is decreased when the monitored processor utilization drops below a second processor utilization threshold lower than ~~the~~ a first processor utilization threshold ~~and the monitored interface buffer utilization drops below a second buffer threshold lower than the first buffer threshold~~ used for identifying when to increase the amount of audio signal encoded in the audio packets.

20. (Currently Amended) A system for throttling network packets in a voice gateway, comprising:

means for encoding an audio signal;

means for formatting the encoded audio signal into Voice over Internet Protocol (VoIP) packets using ~~a~~ multiple central processing ~~unit~~ units;

means for storing the VoIP packets in an interface buffer;

means for monitoring utilization of at least one of the interface buffer and the central processing ~~unit; and~~ units;

means for controlling size of the VoIP packets by increasing a number of samples of the encoded audio signal in the VoIP packets when the monitored utilization indicates high utilization of at least one of the interface buffer and the central processing ~~unit~~ units; and

means for varying a percentage of the multiple central processing units used for increasing the number of samples in the VoIP packets according to the monitored utilization.

21. (Currently Amended) A system according to claim 20 including means for formatting the encoded audio signals using the multiple central processing ~~unit~~ units and varying the VoIP packet size according an amount of processing capacity of the multiple central processing ~~unit~~ units used for formatting the encoded audio signal into VoIP packets.

22. (Previously Presented) A system according to claim 21 including:

means for storing the VoIP packets in the interface buffer before transmitting the VoIP packets over a VoIP network;

means for monitoring the interface buffer by determining an amount of free space in the interface buffer currently not storing VoIP packets; and

means for controlling the VoIP packet size according to the amount of free space currently in the interface buffer.

23. (Currently Amended) A system according to claim 22 including means for periodically monitoring the amount of free space in the interface buffer and the available processing capacity of the multiple central processing ~~unit~~ units and controlling the VoIP

packet size according to that periodic monitoring.

24. (Previously Presented) A system for throttling network packets in a voice gateway, comprising:

means for encoding an audio signal;

means for formatting the encoded audio signal into Voice over Internet Protocol (VoIP) packets using a central processing unit;

means for storing the VoIP packets in an interface buffer;

means for monitoring utilization of at least one of the interface buffer and the central processing unit;

means for controlling size of the VoIP packets by varying a number of samples of the encoded audio signal in the VoIP packets according to the monitored utilization;

means for formatting the encoded audio signals using the central processing unit and varying the VoIP packet size according an amount of processing capacity of the central processing unit used for formatting the encoded audio signal into VoIP packets;

means for storing the VoIP packets in the interface buffer before transmitting the VoIP packets over a VoIP network;

means for monitoring the interface buffer by determining an amount of free space in the interface buffer currently not storing VoIP packets;

means for controlling the VoIP packet size according to the amount of free space currently in the interface buffer; and

means for using multiple digital signal processors to encode multiple audio signals at the same time and varying a percentage of the digital signal processors that increase the VoIP packet size according to the amount of free space in the interface buffer and an amount of processing capacity of the central processing unit used for switching the encoded audio signal to the IP network.

25. (Previously Presented) A system according to claim 20 including:

means for attaching an Internet Protocol header to the encoded audio signal;

means for attaching a User Datagram Protocol (UDP) header to the encoded audio signal; and

means for attaching a Realtime Transport Protocol (RTP) header to the encoded audio signal.

26. (Previously Presented) A system according to claim 20 including means for increasing a number of samples of the audio signal in the VoIP packets when utilization in the interface buffer is above a first threshold and lowering the number of samples of the audio signal samples in the VoIP packets when utilization in the interface buffer drops below a second threshold lower than the first threshold.